VoIP

Voice over IP

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- VoIP aka IP Telephony
- Uses packet-switched network
    - Traditional telephony uses a circuit-switched network, Public Switched Telephone Network (PSTN)
- Can use any IP network
    - Routing of voice conversations over the Internet or any other IP network
    - WAN
    - LAN
- Lower infrastructure costs
    - Only one net

Each service, program, phone, gateway, etc typically supports several different codecs, negotiate which codec to use

- GIPS - Global IP Sound
    13.3 Kbps and up
- GSM
    13 Kbps (full rate), 20ms frame size
- iLBC - internet Low Bitrate Codec
    Free
    Used by Skype.
    15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- ITU G.711 - International Telecommunication Union
    Two flavors: u-law and a-law
    64 Kbps, sample-based Also known as alaw/ulaw
- ITU G.722
    48/56/64 Kbps
- ITU G.723.1
    Used in the H.323 standard
    Requires lisencing by Sipro Lab Telecom
    5.3/6.3 Kbps, 30ms frame size

- ITU G.726
    16/24/32/40 Kbps
- ITU G.728
    16 Kbps
- ITU G.729
    Requires licensing by Sipro Lab Telecom or VoiceAge
    Some free for non-commerical experimental use implementations
    8 Kbps, 10ms frame size
- Speex
    Open-source
    2.15 to 44.2 Kbps
- LPC10 - Linear Predictive Coding
    Voice signal is clear but sounds robotic.
    2.5 Kbps
- DoD CELP
    Also known as Federal Standard 1016
    4.8 Kbps

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The majority of implementations use Real-time Transport Protocol (RTP) to transmit VoIP traffic. The notable exception is IAX which carries both signaling and voice data over a UDP stream (which eases firewall and NAT traversal).

- SIP - Session Initiation Protocol
    An IETF newcomer gaining popularity
    Gradually replacing H.323
- H.323
    The ITU's widely deployed and continually updated VoIP protocol carrying billions of minutes of traffic each month
    Used by NetMeeting and GnomeMeeting
- SCCP - Skinny Client Control Protocol
    Proprietary protocol from Cisco
- Megaco (a.k.a. H.248) and MGCP
    Both media gateway control protocols
- MiNET
    Proprietary protocol from Mitel
- IAX - Inter-Asterisk eXchange
    Used by the Asterisk open-source PBX server and associated client software

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- **Software**
  - Skype
    - Free to use between Skype clients
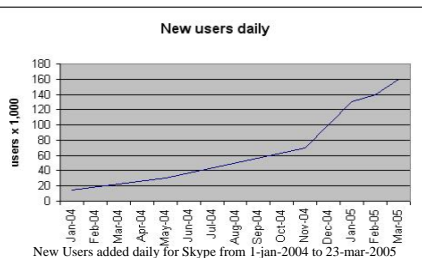- **Hardware (Norway)**
  - Telio - τελιο [telio] (Greek) = perfection
    - IP to analog adapter
    - Monthly fee
    - No per minute fee (Except to cell phones)
  - Telenor - Bredbåndstelefoni
    - IP to analog adapter
    - Monthly fee
    - Free use between Telenor VoIP customers
    - Else per minute fee
  - NextGenTel – NextPhone
  - Tele2 (coming soon)
  - Local providers. El-companies etc...

- **Founded by Niklas Zennström and Janus Friis, the creators of Kazaa**
- **Peer-to-peer**
- **Code and protocol is closed and proprietary**
- **SkypeOut**
  - IP   to PSTN gateway
- **SkypeIn (Beta)**
  - PSTN to IP gateway
- **Skype Voicemail (Beta)**

New users daily

users x 1,000

180
160
140
120
100
80
60
40
20
0

Jan-04 Feb-04 Mar-04 Apr-04 May-04 Jun-04 Jul-04 Aug-04 Sep-04 Oct-04 Nov-04 Dec-04 Jan-05 Feb-05 Mar-05

New Users added daily for Skype from 1-jan-2004 to 23-mar-2005

- http://www.voip-info.org/
- http://en.wikipedia.org/wiki/VoIP
- http://www.skype.com/
- http://en.wikipedia.org/wiki/Skype
- http://www.telio.no/
- http://privat.telenor.no/telefoni/default.aspx?a=goProduktBBT
- http://www.nextgentel.no/privat/produkter/nextphone/
- http://www.tele2.no/privat/bredbandstelefoni/